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APPLICATION NO.	FILING DATE	FIRST NAMED INVENTOR	ATTORNEY DOCKET NO.	CONFIRMATION NO.
09/849,719	05/04/2001	Kenneth H.P. Chang	SSI001US	9646

27906 7590 02/16/2005

PATENT LAW OFFICES OF DAVID MILLERS
6560 ASHFIELD COURT
SAN JOSE, CA 95120

EXAMINER	
VO, HUYEN X	
ART UNIT	**PAPER NUMBER**
2655	

DATE MAILED: 02/16/2005

Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)	
	09/849,719	CHANG, KENNETH H.P.	
	Examiner	**Art Unit**	
	Huyen Vo	2655	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on _29 October 2004_.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-25_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-25_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _04 May 2001_ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Response to Arguments

1. Applicant's arguments, see amendment, filed 10/29/2004, with respect to the rejection(s)of claim(s) 1-26 under 35 USC 102 and 103 have been fully considered and are persuasive. Therefore, the rejection has been withdrawn. However, upon further consideration, a new ground(s) of rejection is made in view of Taniguchi et al. (US 6484137) and Gupta et al. (US 6622171).

Claim Rejections - 35 USC § 102

1. The following is a quotation of the appropriate paragraphs of 35 U.S.C. 102 that form the basis for the rejections under this section made in this Office action:

> A person shall be entitled to a patent unless – (e) the invention was described in (1) an application for patent, published under section 122(b), by another filed in the United States before the invention by the applicant for patent or (2) a patent granted on an application for patent by another filed in the United States before the invention by the applicant for patent, except that an international application filed under the treaty defined in section 351(a) shall have the effects for purposes of this subsection of an application filed in the United States only if the international application designated the United States and was published under Article 21(2) of such treaty in the English language.

2. Claims 12-13 and 19-21 are rejected under 35 U.S.C. 102(e) as being anticipated by Taniguchi et al. (US 6484137).

3. Regarding claim 12, Taniguchi et al. disclose an apparatus containing a data structure representing an audio presentation, the data structure comprising a plurality of audio channels representing the audio presentation after time scaling, wherein: each audio channel has a corresponding time scale factor and includes a plurality of audio frames (*col. 14, lines 7-67, where "a"=1 and "b"=1/2 or referring to figure 4c for more*

details); and each audio frame has a frame index that uniquely distinguishes the audio frame from other audio frames in the same channel and identifies the audio frame as corresponding to specific audio frames in other audio channels (*col. 28, lines 30-65*).

4. Regarding claim 13, Taniguchi et al. further disclose the apparatus of claim 12, wherein audio frames that are in different channels and have the same frame index represent the same portion of the audio presentation (*col. 28, lines 30-65*).

5. Regarding claim 19, Taniguchi et al. disclose a method for playing a presentation, comprising: loading a first frame from a source into a player via a network, the first frame representing a first portion of the presentation after scaling by a first time-scaling factor (*col. 14, lines 7-64*), the first audio frame has a first channel index value that identifies the first audio frame as being scaled by the first time scaling factor (*col. 14, lines 7-49 or col. 27, lines 58-65*); playing the first audio frame to provide the first portion of the presentation with the first time scale factor (*col. 14, lines 7-64*); receiving a request to change playing from the first time scaling factor to a second time scaling factor (*col. 15, lines 3-5*); requesting from the source a second audio frame that has a second channel index value that identifies the second audio frame as being scaled by the second time-scaling factor (*col. 15, line 6 to col. 16, line 20*); and playing the second audio frame after the first audio frame to provide a real-time change in the time-scale of the presentation (*col. 15, line 6 to col. 16, line 20*).

6. Regarding claims 20-21, Taniguchi et al. further disclsoe that the first frame has a first frame index value that identifies the first portion of the presentation that the first audio frame represents, and the second frame has a second index value that identifies a second portion of the presentation that the first audio frame represents (*col. 14, lines 7-49 or col. 27, lines 58-65*); and the second index value immediately follows the first time index value (*col. 14, lines 7-49 or col. 27, lines 58-65*).

7. Claims 1-5, 9-11, 14-18, and 24-25 are rejected under 35 U.S.C. 102(e) as being anticipated by Gupta et al. (US 6622171).

8. Regarding claim 1, Gupta et al. disclose an apparatus containing a data structure representing a presentation, the data structure comprising: a first audio channel representing an audio portion of the presentation after time scaling by a first time scale factor (*referring to figure 8 and/or col. 11, lines 1-67*); and a second audio channel representing the audio portion after time scaling by a second time scale factor that differs from the first time scale factor (*referring to figure 8 and/or col. 11, lines 1-67*).

9. Regarding claim 14, Gupta et al. disclose a method for encoding audio data, comprising: performing a plurality of time scaling processes on the audio data to generate a plurality of time-scaled audio data sets, each time-scaled audio data set having a different time scale factor (*figure 8 shows 3 versions of an audio signal being scaled by 3 different timing scale factor*); and generating a data structure containing a plurality of audio channels respectively corresponding to the plurality of time scaling

processes (*figure 8*), wherein content of each of the audio channels is derived from the time-scaled audio data set resulting from performing the corresponding time scaling process on the audio data (*figure 8*).

10. Regarding claims 2-3, 5, and 9, Gupta et al. further disclose that the first audio channel comprises plurality of frames (*figure 8, audio signal is processed frame by frame, known in the art*); the second audio channel comprises plurality of frames that are in one-to-one correspondence with the plurality of frames in the first audio channel (*figure 8 represents 3 versions of an audio signal after being scaled by 3 different time scale factor*); and corresponding frames in the first and second audio channels represent the same time interval of the presentation (*figure 8, frames in each version of scaled audio signals shown in figure 8 are corresponding with each other*), and each frame in the first audio channel is separately compressed using a first compression method (*each frame of an audio signal is compressed on the one-by-one basis*), and wherein the data structure further comprises a data channel identifying graphics associated with the audio presentation (*col. 11, lines 44-53, selection appropriate version of the video stream to combine with audio stream*), and wherein the apparatus comprises a server connected to a network (*figures 8-9*).

11. Regarding claims 10-11, Gupta et al. further disclose that the apparatus comprises: data storage in which the data structure is stored (*figure 1, particularly element 13*); a decoder connected to receive a data stream, the decoder converting the data stream for perceivable presentation (*Decoders 108-109 in figure 3*); and selection

logic coupled to the data storage and capable of selecting a source channel for the data stream from among a set of channels including the first audio channel and the second audio channel (*element 130-140 in figure 4*), wherein the apparatus is a standalone device that operates on battery power (*client device 11 in figure 1*).

12. Regarding claim 4, Gupta et al. further disclose that the data structure further comprises a third audio channel representing the audio presentation after time scaling by the first time scale factor, wherein each frame in the third audio channel is separately compressed using a second compression method (*figure 8 shows 3 versions of an audio signal that are time scaled by 3 different time scale factor, wherein each represent an individual audio channel*).

13. Regarding claims 15-18, Gupta et al. further disclose that generating the data structure comprises: partitioning each time-scaled audio data set into a plurality of frames (*figure 3, multimedia data transmitted to the client device in packets or frames*); separately compressing each frame to produce compressed frames (*since the client device 11 includes audio/video decoders, the servers must have compressed the data frame by frame before transmitting to the client device*); and collecting the compressed frames into the plurality of audio channels, each audio channel having a corresponding one of the different time scale factors (*figure 8 shows 3 versions of an audio signal being scaled by three different time scale factor*), wherein all frames resulting from partitioning correspond to the same amount of time in the audio data (*fixed frame size,*

known in the art), wherein separately compressing each frame comprises applying a plurality of different compression processes to generate a plurality of compressed frames from each frame (*figure 8 shows 3 time-scaled compression versions of an audio signal*); and collecting the compressed frames produces audio channels such that in each audio channel, all compressed frames in the audio channel have the same time scale and compression process (*figure 8 shows 3 time-scaled compression versions of an audio signal*).

14. Regarding claim 24, Gupta et al. discloses a method for playing an audio presentation on a receiver that is connected via a network to a source having a multi-channel data structure representing the audio presentation, the method comprising: determining available bandwidth on the network (*col. 11, lines 44-53*); selecting a first channel of the multi-channel data structure from a plurality of channels that represent the audio presentation after time-scaling by a desired time-scaling factor (*col. 11, line 44 to col. 12, line 67*); receiving a first frame from the first channel (*col. 11, line 44 to col. 12, line 67*); and playing the first frame (*col. 11, line 44 to col. 12, line 67*).

15. Regarding claim 25, Gupta et al. further disclose that determining bandwidth available on the network after receiving the first frame (*col. 8, line 19 to col. 9, line 67*); selecting a second channel of the multi-channel data structure from the plurality of channels that represent the audio presentation after time-scaling by the desired time-scaling factor (*col. 11, line 44 to col. 12, line 67*), wherein the second channel contains

data that is compressed using a second compression process that provides highest audio quality at the bandwidth available after receiving the first frame; receiving a second frame from the second channel (*col. 11, line 44 to col. 12, line 67*); and playing the second frame after playing the first frame (*col. 11, line 44 to col. 12, line 67*).

Claim Rejections - 35 USC § 103

16. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

17. Claims 6-8 are rejected under 35 U.S.C. 103(a) as being unpatentable over Gupta et al. (US 6622171) in view of Taniguchi et al. (US 6484137).

1. Regarding claim 6, Gupta et al. fail to disclose that the first audio channel comprises plurality of frames, each frame having an index value that identifies a time interval of the audio portion that the frame represents; the second audio channel comprises plurality of frames, each frame in the second channel having an index value that identifies a time interval of the audio portion that the frame represents. However, Taniguchi et al. teach that the first audio channel comprises plurality of frames, each frame having an index value that identifies a time interval of the audio portion that the frame represents (*col. 28, lines 30-65*); the second audio channel comprises plurality of

frames, each frame in the second channel having an index value that identifies a time interval of the audio portion that the frame represents (*col. 28, lines 30-65*).

Since Gupta et al. and Taniguchi et al. are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Gupta et al. by incorporating the teaching of Taniguchi et al. in order to determine data blocks [redacted] subjected to time-scale modification to [redacted] minimize audio distortion.

18. Regarding claims 7-8, Gupta et al. further disclose that each frame in the first and second data channels is separately compressed (*figure 8 show 3 versions of an audio stream being encoded by 3 different time scale factors*), and wherein the data structure further comprises a data channel corresponding to a plurality of bookmarks, wherein each bookmark has index value and identifies graphics, the index value indicating a display time for the graphics relative to playing of the frames of the first or second audio channel (*col. 4, lines 1-20*).

19. Claims 22-23 are rejected under 35 U.S.C. 103(a) as being unpatentable over Taniguchi et al. (US 6484137) in view of Gupta et al. (US 6622171).

20. Regarding claim 22, Taniguchi et al. further disclose that the channel index values of frames further indicate respective compression processes for the frames, and wherein the method further comprises: selecting the second channel index value (*col.*

28, lines 51-65) from a plurality of channel index values that identify the second time scaling factor (*col. 15, line 3 to col. 16, line 19*).

Taniguchi et al. fail to disclose the steps of determining available bandwidth on the network, and wherein the second channel index indicates a compression process provides highest audio quality at the available bandwidth. However, Gupta et al. teach the steps of determining available bandwidth on the network (*col. 11, lines 44-53*), and wherein the second channel index indicates a compression process that provides highest audio quality at the available bandwidth (*col. 11, line 44 to col. 12, line 67*).

Since Taniguchi et al. and Gupta et al are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Taniguchi et al. by incorporating the teaching of Gupta et al. in order to determine data blocks ~~to be~~ subjected to time-scale modification to ~~that~~ minimize audio distortion.

21. Regarding claim 23, Taniguchi et al. further disclose that the channel index values of frames further indicate respective compression processes for the frames, and wherein the method further comprises: selecting a third channel index value from a plurality of channel index values that identify the second time scaling factor (*col. 14, line 7 to col. 16, line 19*), requesting from the source a third audio frame that has the third channel index value, which identifies the third audio frame as being time-scaled by the second time-scaling factor (*each frame in the frame sequence in col. 12, lines 12-13 has a different time-scaling factor, e.g. "a"=1 and "b"=1/2, for more detail, referring to*

figure 4c); and playing the third frame after the second frame (*col. 14, line 7 to col. 16, line 19*).

Taniguchi et al. fail to disclose the steps of determining available bandwidth on the network, wherein the third channel index indicates a compression process that provides highest audio quality at the available bandwidth. However, Gupta et al. teach the steps of determining available bandwidth on the network (*col. 11, lines 44-53*), and wherein the third channel index indicates a compression process that provides highest audio quality at the available bandwidth (*col. 11, line 44 to col. 12, line 67*).

Since Taniguchi et al. and Gupta et al are analogous art because they are from the same field of endeavors, it would have been obvious to one of ordinary skill in the art at the time of invention to modify Taniguchi et al. by incorporating the teaching of Gupta et al. in order to determine data blocks [redacted] subjected to time-scale modification to [redacted] minimize audio distortion.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Huyen Vo whose telephone number is 703-305-8665. The examiner can normally be reached on M-F, 9-5:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Doris To can be reached on 703-305-4827. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

Huyen X. Vo February 15, 2005

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Susan McFadden

SUSAN MCFADDEN
PRIMARY EXAMINER